Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The objective of the compliance test was to validate interoperability of LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal.

LumenVox provides a complete set of speech recognition, text-to-speech technologies, and call progress analysis for use in interactive voice response (IVR) applications. The product set includes the LumenVox Automatic Speech Recognizer (ASR), Text-to-Speech (TTS) Server, and Call Progress Analysis (CPA). All products are used in conjunction with the LumenVox Media Server which provides an interface to Avaya Aura® Experience Portal using the Media Resource Control Protocol (MRCP). Additionally LumenVox Call Progress Analysis (CPA) leverages the strength of LumenVox Automated Speech Recognizer (ASR) by constantly listening for various tones, just as it would when performing speech recognition. These are compared to special acoustic models for matches, similar to our ASR's function. The result is more reliable and highly accurate message delivery.

2. General Test Approach and Test Results

The general test approach was to test various VoiceXML scripts that exercise various types of grammars in LumenVox ASR and TTS. A predefined set of VoiceXML scripts tested built-in grammars, menu grammars and Speech Recognition Grammar Specification (SRGS) grammars. Verification of CPA was performed using outbound calls for testing CPA capabilities.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability tests. Feature tests focused on the ability of LumenVox ASR and TTS to successfully exercise appropriate grammar and return expected results. LumenVox CPA focused on proper detection of answering party.

Serviceability testing focused on verifying the ability of the LumenVox ASR, TTS, and CPA server to recover from adverse conditions, such as restart, power failures and network disconnects.

2.2. Test Results

All test cases passed.

2.3. Support

To obtain technical support for LumenVox:

- **Web:** www.lumenvox.com/help/
- **Email:** support@lumenvox.com
- **Phone:** (858)707–7700

3. Reference Configuration

Following diagram shows the configuration used during the interoperability compliance test. Reference configuration consisted of:

- Avaya Aura® Experience Portal
- Avaya S8300D Server running Avaya Aura® Communication Manager
- Avaya G450 Media Gateway
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- Avaya IP Telephones
- Application Server
- LumenVox Automated Speech Recognizer
- LumenVox Text-to-Speech Server
- LumenVox CPA

Note: Each of the LumenVox components are installed on a single server.

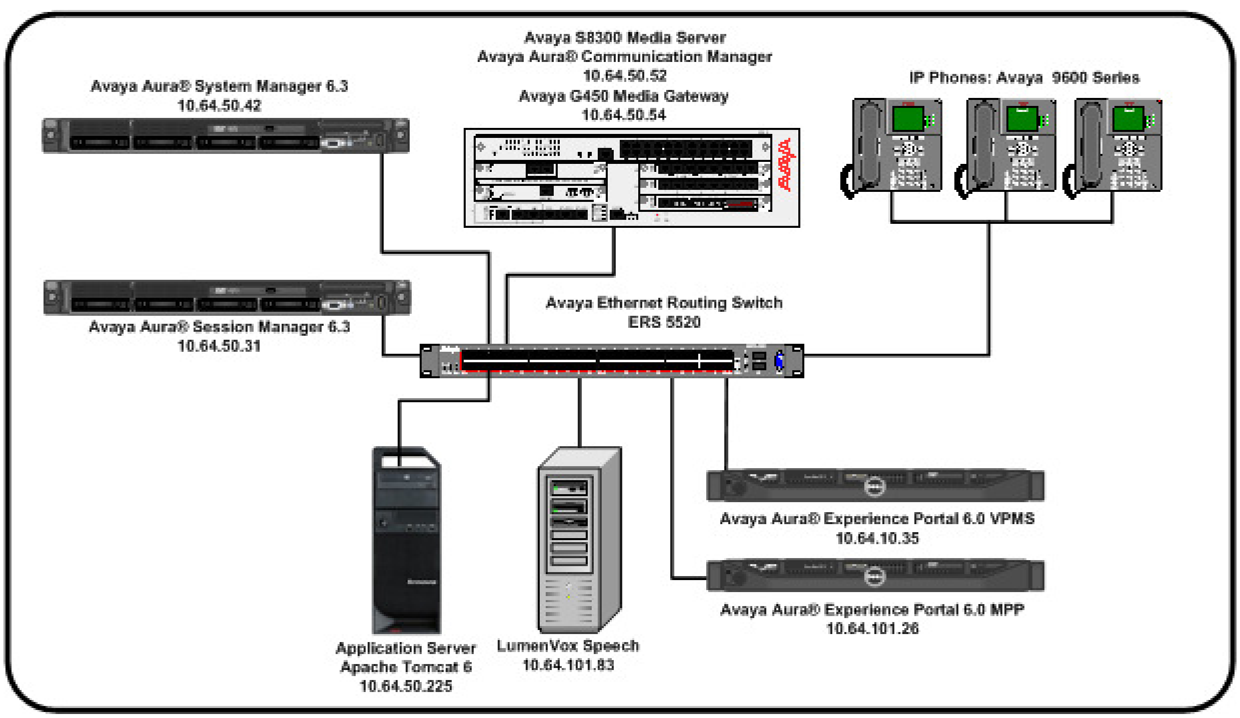


Figure 1: Reference Configuration

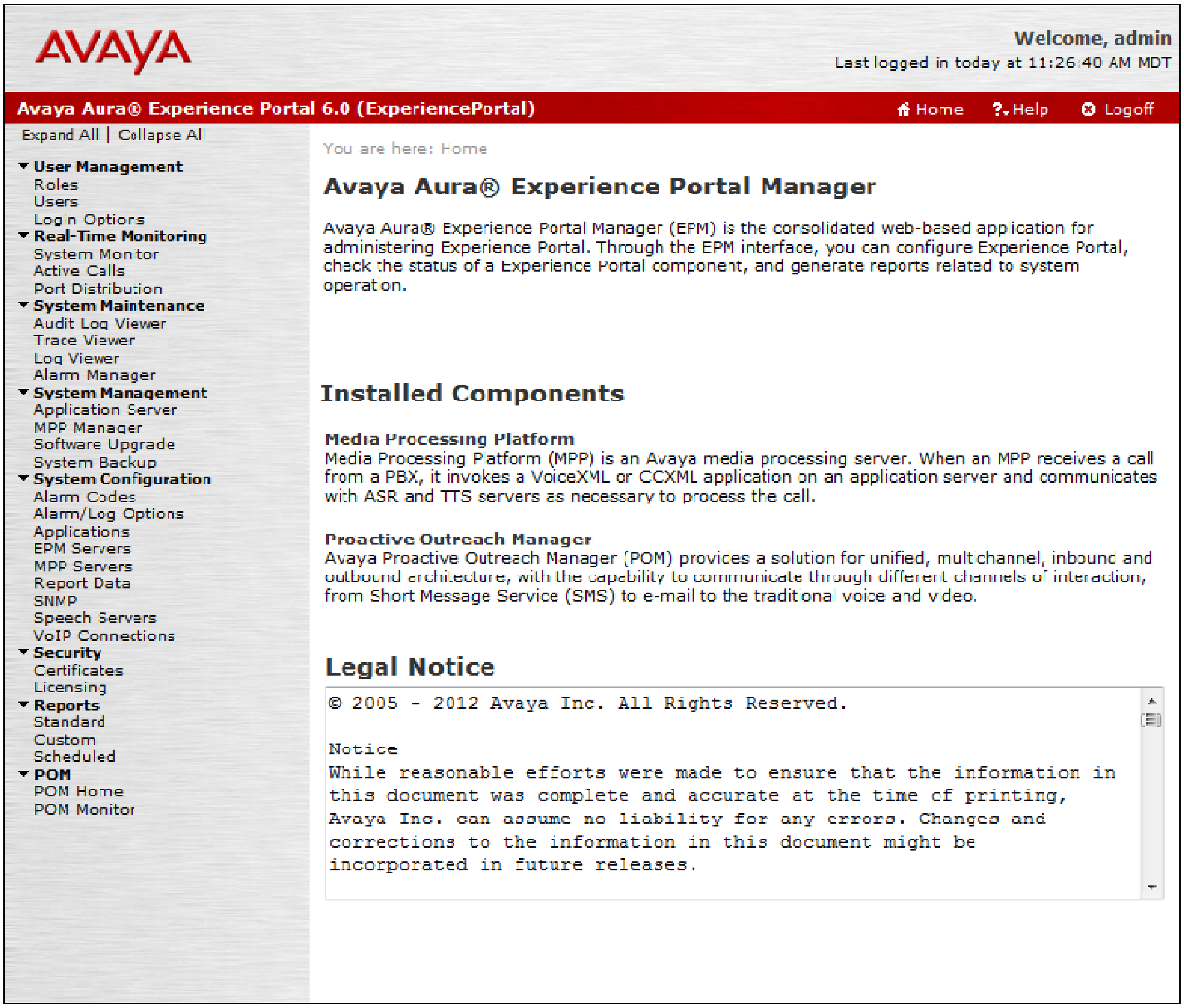
4. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Equipment/Software	Release/Version
Avaya Aura® Experience Portal EPM running on HP Proliant DL360 G7	6.0.2.0.0501
Avaya Aura® Experience Portal MPP running on VM Ware Virtual Machine on Dell Blade Server	6.0.2.0.0501
Avaya G450 Media Gateway	32.26.0
Avaya S8300D Server running on Avaya Aura® Communication Manager	R016x.03.0.124.0
Avaya 9600 Series IP Telephones	H.323 3.2 SIP 6.2.2.17
Avaya Aura® Session Manager running on HP Proliant DL360 G7	6.3.2.0.632023
Avaya Aura® System Manager running on HP Proliant DL360 G7	6.3.0 –FP2
LumenVox components on a single VM Ware Virutal Machine running on Dell Blade Server: • LumenVox Automated Speech Recognizer • LumenVox Text-to-Speech • LumenVox Call Progress Analysis	11.2.200
Tomcat Apache Web Server running on VM Ware Virtual Machine running on Dell Blade Server	6.0.29

5. Configure Avaya Aura® Experience Portal

Experience Portal is configured via the Experience Portal Manager (EPM) web interface. To access the web interface, enter http://<*ip-addr*>/ as the URL in a web browser, where <*ip-addr*> is the IP address of the EPM. Log in using the admin user. (Not Shown)



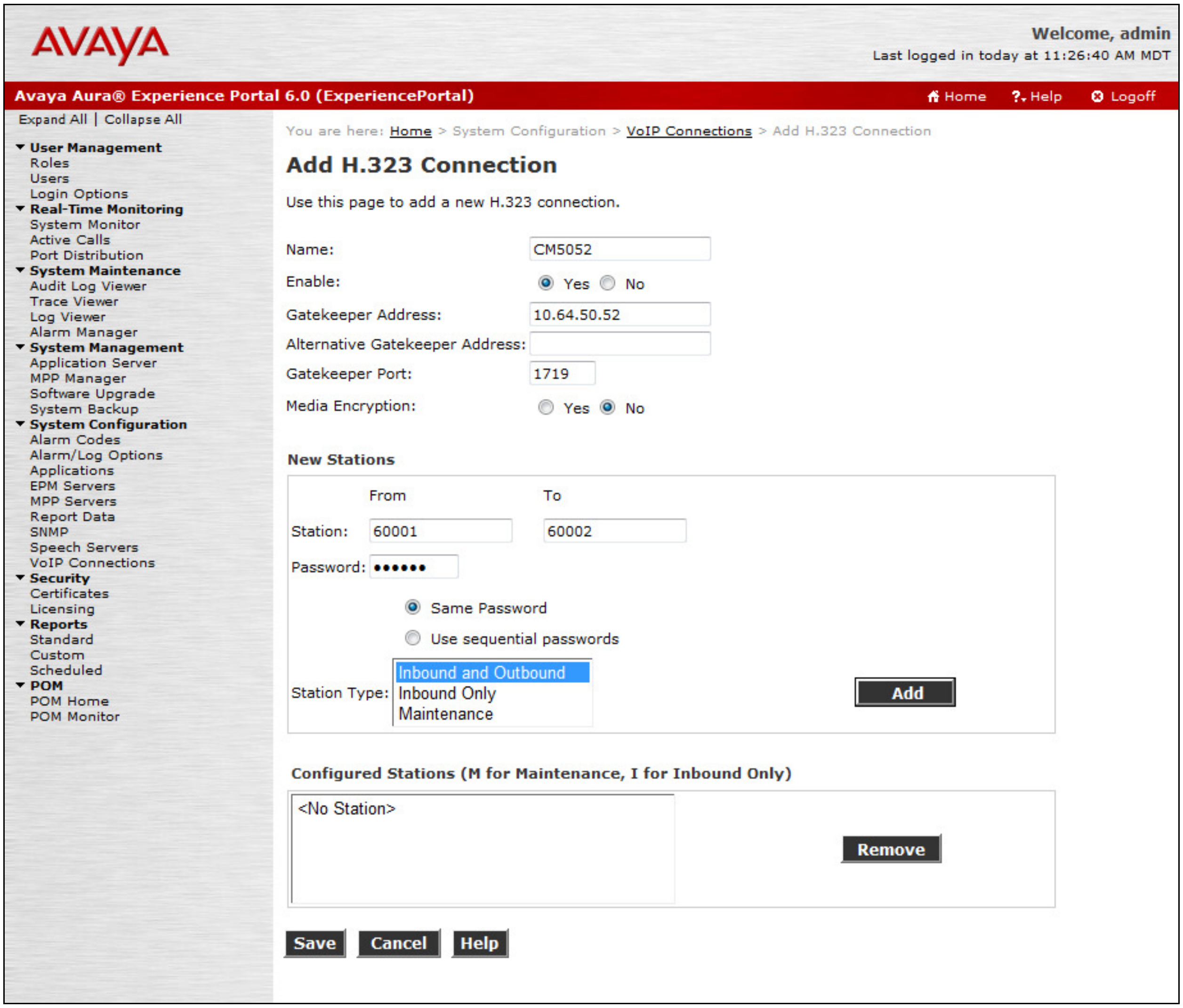
5.1. Add VoIP Connections

During compliance testing both H.323 and SIP VoIP connections were used. However, either H.323 or SIP VoIP connection can be active at any given time. On the left pane, click on the **System Configuration →VoIP Connections** tab to configure VoIP connections (not shown).

5.1.1. H.323 Connection

To add an H.323 Connection, click on **H.323** tab (not shown) and click **Add** (not shown)

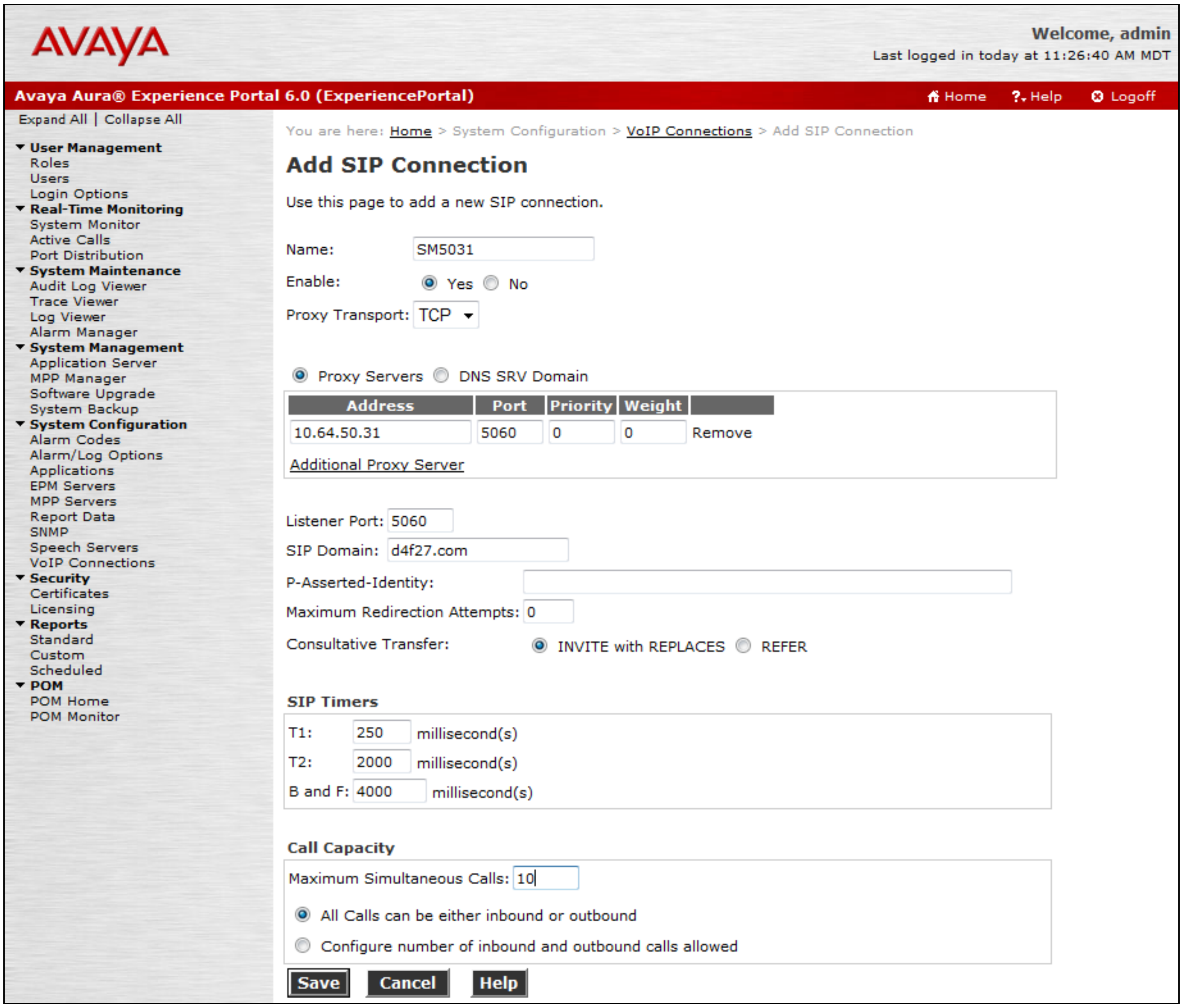
- **Name**: Enter a descriptive name
- **Gatekeeper Address**: Enter the IP address of Communication Manager.
- **Media Encryption**: Set to **No**.
- **New Stations**: Enter **Station From** and **To**, and **Password**. Select **Inbound and Outbound** and click **Add.**
  Note: Station information should be gathered from the existing Communication Manager configuration.
- Retain the default values in the remaining fields Click **Save** to save changes.



5.1.2. SIP Connection

To add a **SIP Connection**, click on the **SIP** tab (not shown) on the **VoIP Connections** page (not shown).

- **Name**: Enter a descriptive name..
- Set **Proxy Transport** to **TCP**.
- In the **Address** and **Port** boxes, enter the IP address and Port of Session Manager.
- **SIP Domain**: Enter the domain used in Session Manager.
- **Maximum Simultaneous Calls**: During the test, **10** was used for the Maximum Simulataneous Calls field.
- Retain the default values in the remaining fields Click **Save** to save changes.



5.2. Add Speech Servers

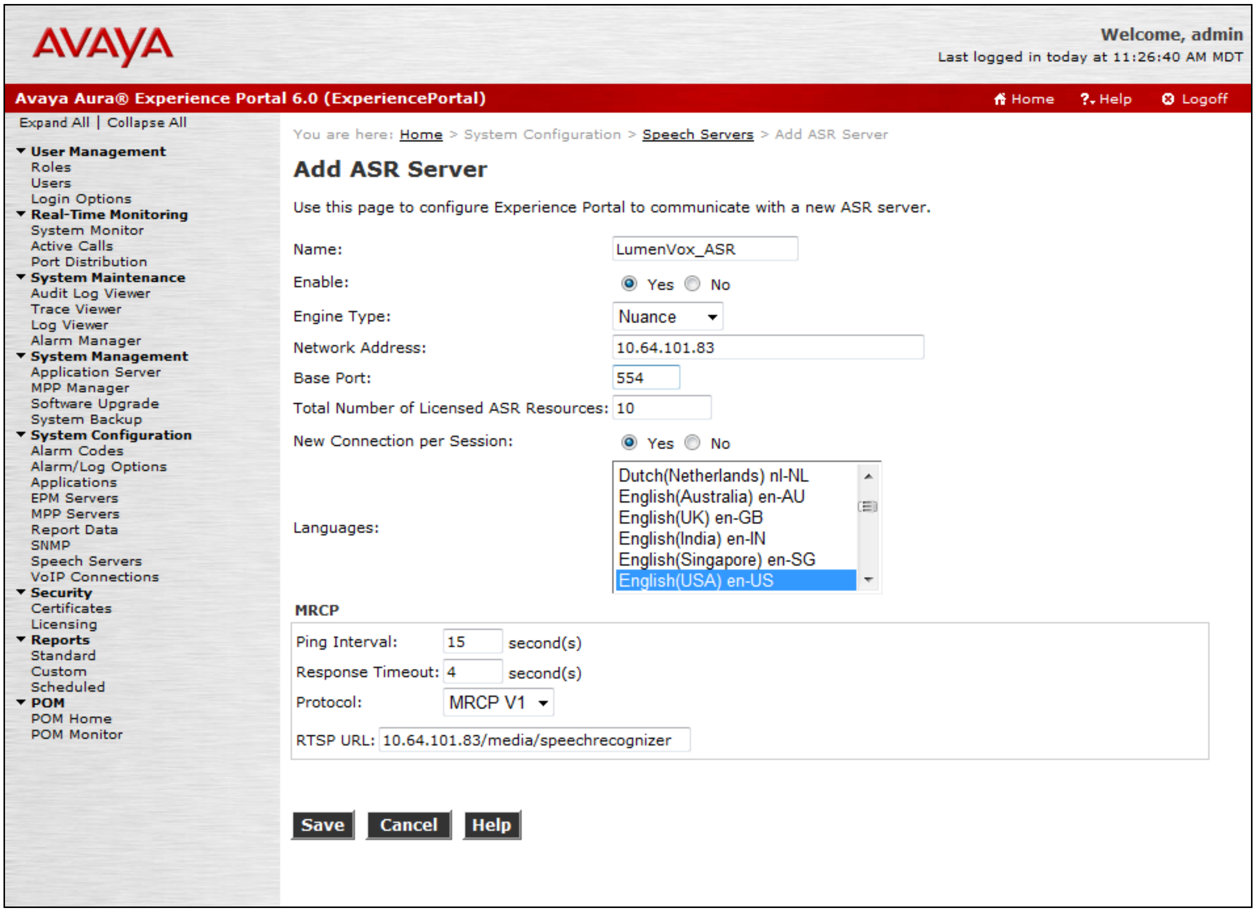
On the left pane, click on the **System Configuration →Speech Servers** tab to add Speech Server.

5.2.1. ASR Server

To add an **ASR** server, click on **ASR** tab (not shown), and click **Add** (not shown).

- **Name**: Enter a descriptive name**.**
- **Enable**: Set to **Yes.**
- **Engine Type**: Set to **Nuance**, using the drop down menu
- **Network Address**: Enter the IP address of LumenVox Automated Speech Recognizer**.**
- **Base Port**: Enter **554.**
- **Total Number of Licensed ASR Resources**: Enter an appropriate value
- **New Connection per Session**: Select **Yes.**
- **Languages**: Select **English(USA) en-US.**

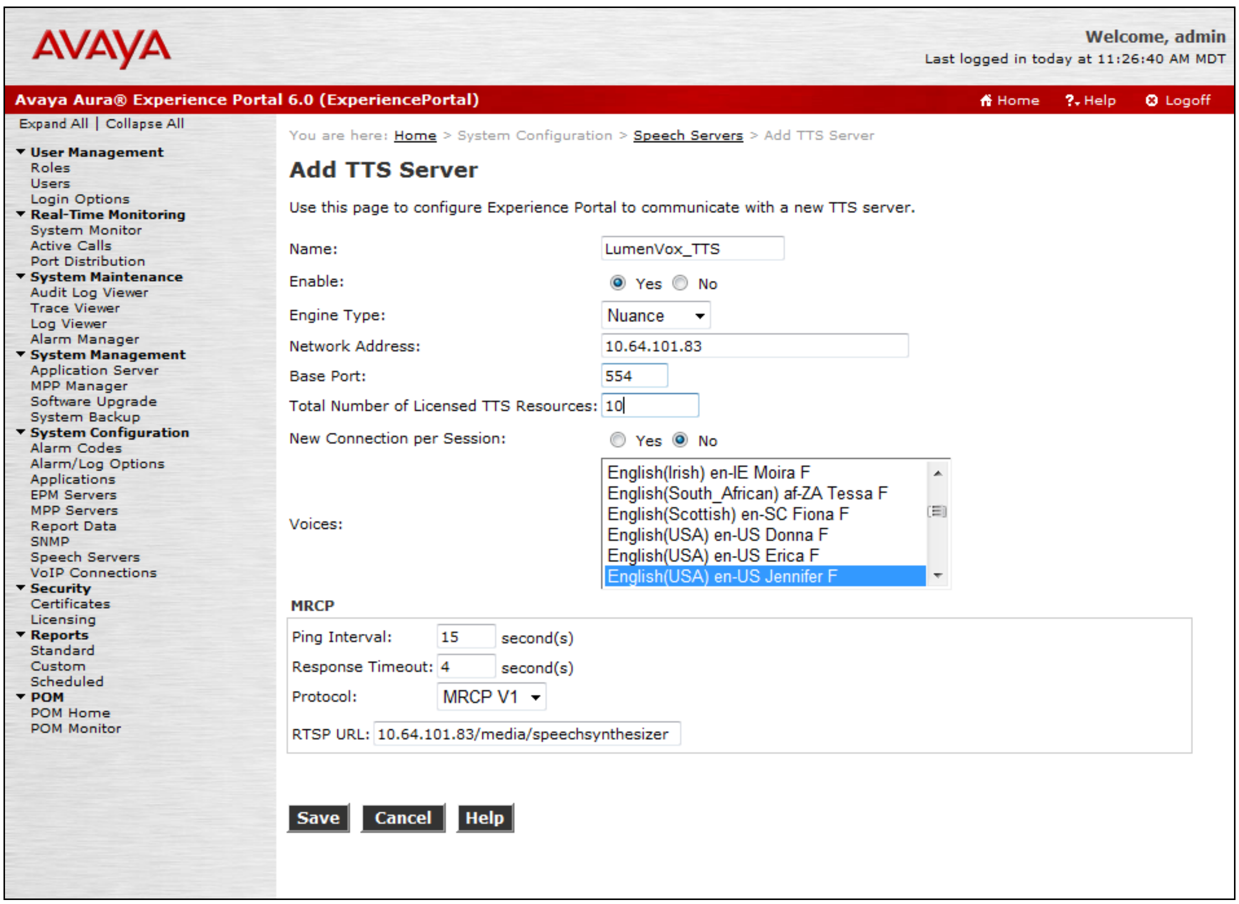
- **RTSP URL**: Enter**<LumenVox_ASR_IP address>/media/speechrecognizer**
- Click **Save** to save changes.



5.2.2. TTS Server

To add a **TTS** server, click on **TTS** tab (not shown) on **Speech Servers** (not shown) page, and click **Add** (not shown)**.**

- **Name**: Enter a descriptive name.
- **Enable**: Set to **Yes.**
- **Engine Type**: Set to **Nuance** using the drop down menu.
- **Network Address**: Enter the IP address of LumenVox Text to Speech server.
- **Base Port**: Enter **554.**
- **Total Number of Licensed TTS Resources**: Enter an appropriate value.
- **New Connection per Session**: Select **Yes.**
- **Languages**: Select **English(USA) en-US Jennifer F.**
- **RTSP URL**: Enter **<LumenVox_ASR_IP address>/media/speechsynthesizer**
- Click **Save** to save changes.



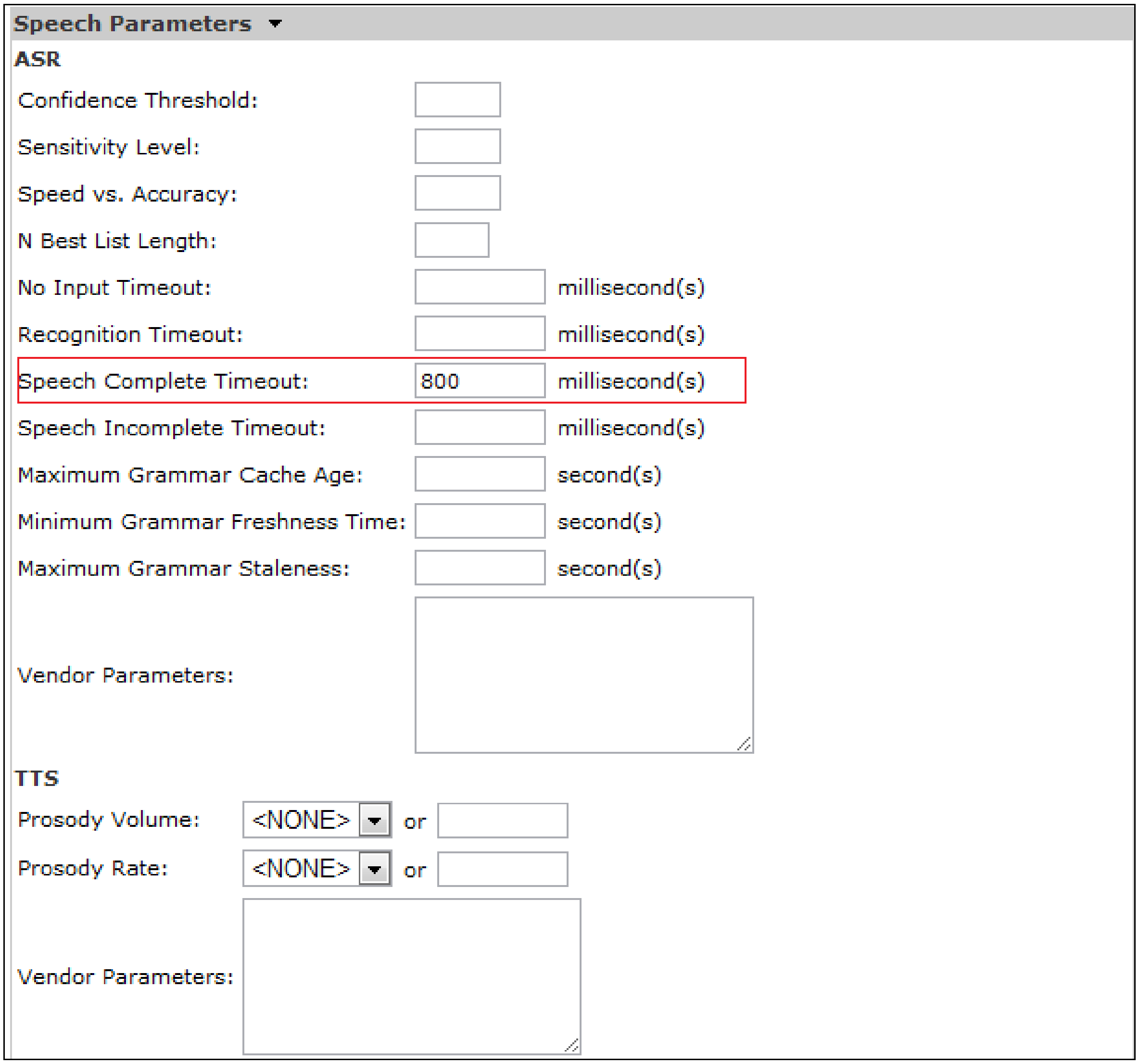
6. Configure LumenVox Automated Speech Recognizer

All configurations for LumenVox applications were performed by a LumenVox Engineer.

Log on to LumenVox server using a SSH client. The **/etc/lumenvox/media_server.conf** file needs to be modified for the following fields:

- The value of **mrcp_server_ip** must be set to the IP address of the machine that LumenVox is installed on. This must be an IP address that the Experience Portal can reach and route traffic to. Please contact LumenVox Support for questions about configuring firewalls if they will be running.
- The value of **compatibility_mode** must be changed from the default **0** to **1**

Note: When configuring an application in Experience portal to use the LumenVox ASR, set the "**Speech Complete Timeout**" parameter under **Speech Parameters** to a non-0 value:



7. Configure LumenVox Text-to-Speech Server

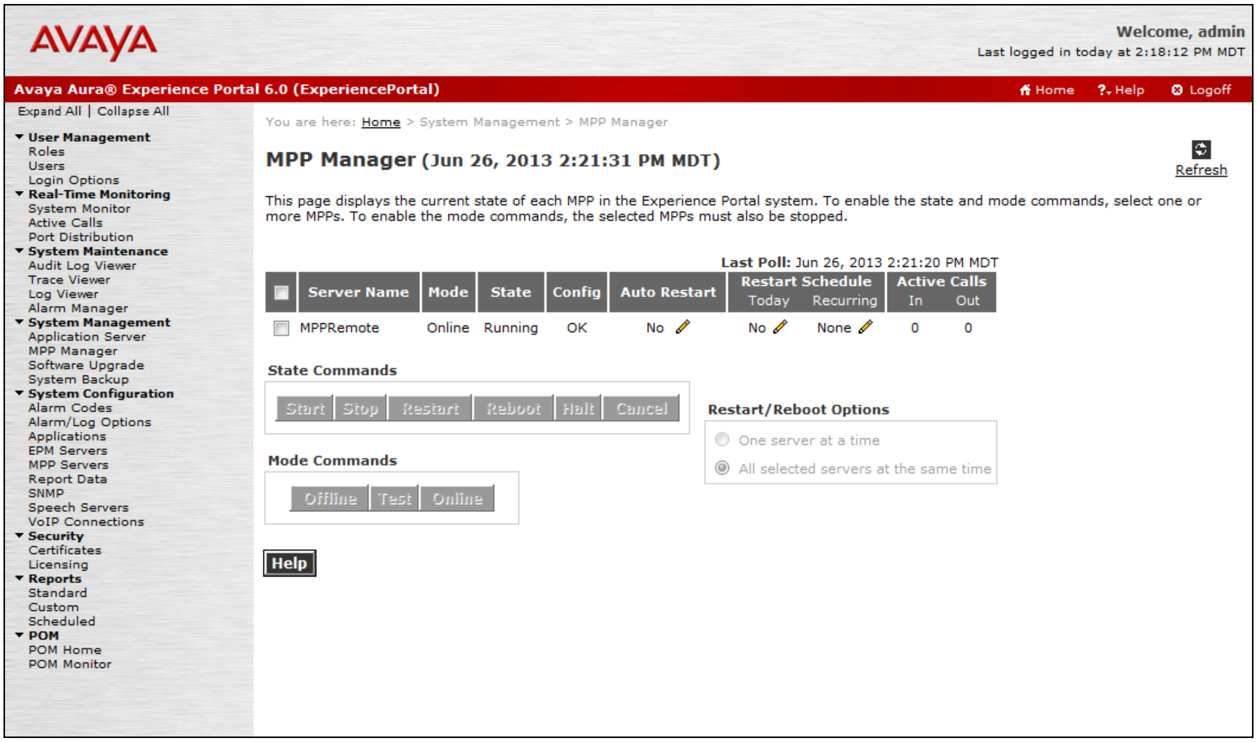
The LumenVox Media Server must be configured as described in **Section 6**. There are no special configurations for the Text-to-Speech Server.

8. Verification Steps

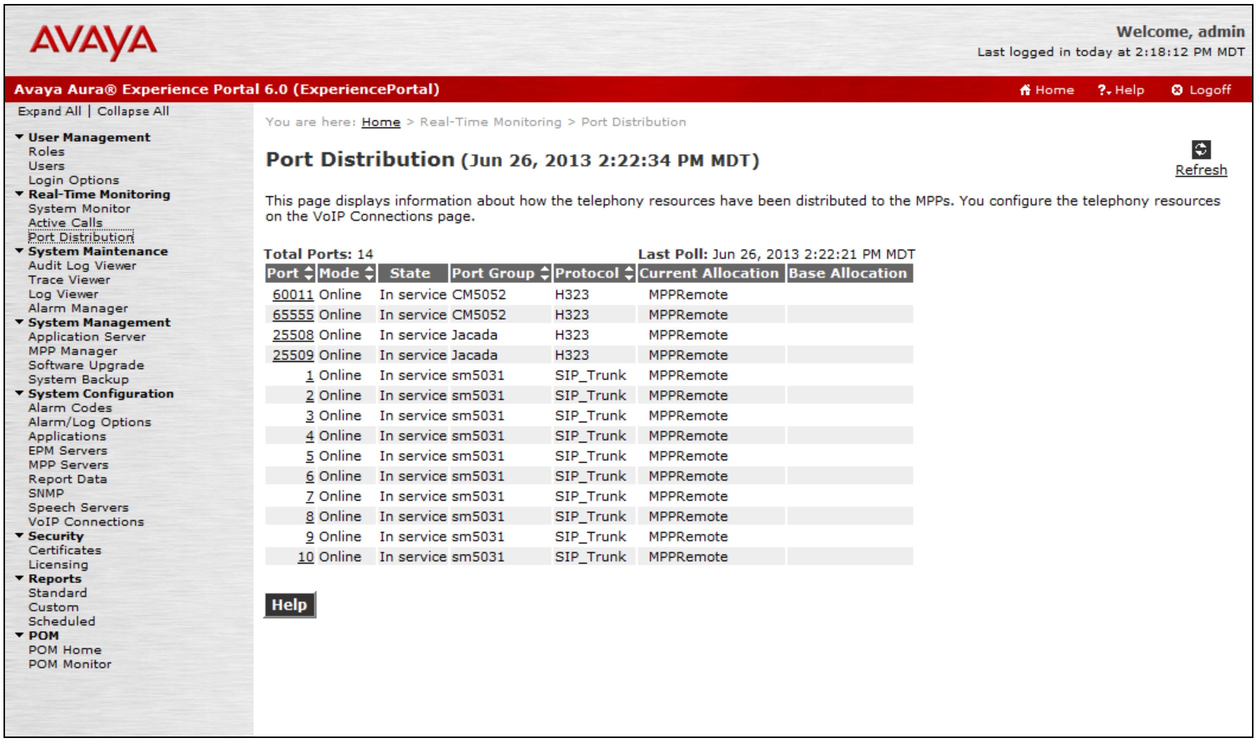
This section provides the verification steps that may be performed to verify that Experience Portal and the LumenVox servers are online and functioning properly.

8.1. Avaya Aura® Experience Portal

1. From the EPM web interface, navigate to **System Management → MPP Manager**. From the MPP Manager screen, shown below, verify that the Media Processing Platform (MPP) servers are **Online** and **Running**.


AVAYA
Welcome, admin
Last logged in today at 2:18:12 PM MDT
Avaya Aura® Experience Portal 6.0 (ExperiencePortal)
Home
Help
Logoff
Expand All | Collapse All
User Management
Roles
Users
Login Options
Real-Time Monitoring
System Monitor
Active Calls
Port Distribution
System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager
System Management
Application Server
MPP Manager
Software Upgrade
System Backup
System Configuration
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections
Security
Certificates
Licensing
Reports
Standard
Custom
Scheduled
POM
POM Home
POM Monitor
You are here: Home > System Management > MPP Manager
MPP Manager (Jun 26, 2013 2:21:31 PM MDT)
Refresh
This page displays the current state of each MPP in the Experience Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.
Last Poll: Jun 26, 2013 2:21:20 PM MDT
Server Name
Mode
State
Config
Auto Restart
Restart Schedule
Today
Recurring
Active Calls
In
Out
MPPRemote
Online
Running
OK
No
No
None
0
0
State Commands
Start
Stop
Restart
Reboot
Halt
Cancel
Restart/Reboot Options
One server at a time
All selected servers at the same time
Mode Commands
Offline
Test
Online
Help

2. On the left pane, navigate to **Real-Time Monitoring → Port Distribution**. From the Port Distribution page, verify that the ports on the MPP server are in service



8.2. LumenVox Automated Speech Recognizer

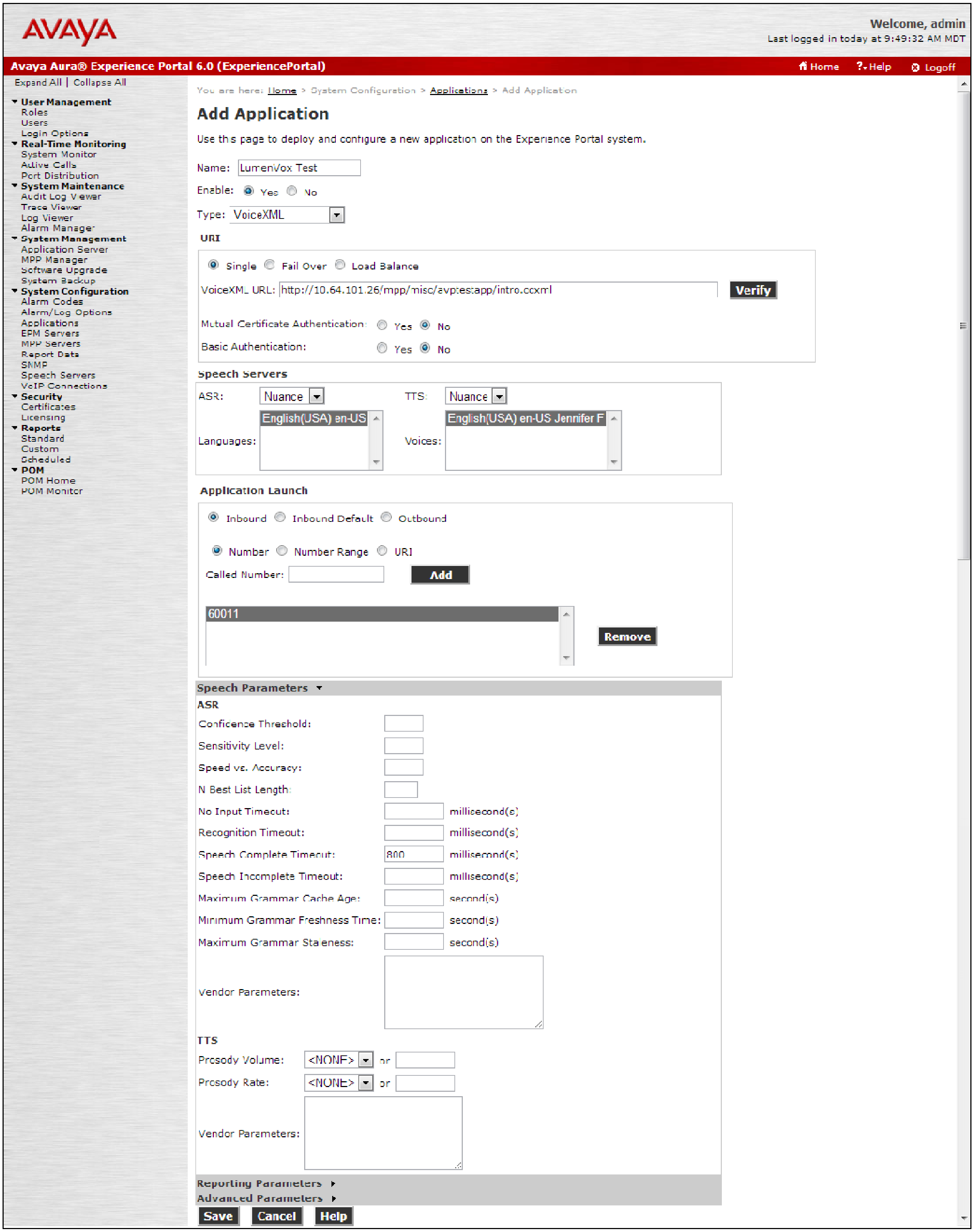
The Avaya test application (usually installed with the MPP in **/mpp/misc/avptestapp/intro.vxml**) may be used to test the ASR.

8.2.1. Configuring the Application

From the EPM web interface, navigate to **System Configuration → Applications**. Click the **Add** button (not shown) to create a new application.

- **Name**: Enter a descriptive name.
- **Enable**: Set to **Yes.**
- **Type**: Set to **VoiceXML** using the drop down menu.
- **URL** Enter the URL for the Application.
- **Speech Servers** Select **Nuance** for **ASR** and **TTS.**
- **Application Launch** Enter an Extension for **Called Number:** and click the **Add** button.
- **Speech Parameters** Enter **800** for **Speech Complete Timeout:**
- Click **Save** to save changes.

  Note: When configuring an application in Experience portal to use the LumenVox ASR, set the "**Speech Complete Timeout**" parameter under **Speech Parameters** to a non-0 value:


AVAYA
Welcome, admin
Last logged in today at 9:49:32 AM MDT
Avaya Aura® Experience Portal 6.0 (ExperiencePortal)
Home
Help
Logoff
Expand All | Collapse All
User Management
Roles
Users
Login Options
Real-Time Monitoring
System Monitor
Active Calls
Port Distribution
System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager
System Management
Application Server
MPP Manager
Software Upgrade
System Backup
System Configuration
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections
Security
Certificates
Licensing
Reports
Standard
Custom
Scheduled
POM
POM Home
POM Monitor
You are here: Home > System Configuration > Applications > Add Application
Add Application
Use this page to deploy and configure a new application on the Experience Portal system.
Name: LumenVox Test
Enable: Yes No
Type: VoiceXML
URI
Single Fail Over Load Balance
VoiceXML URL: http://10.64.101.26/mpp/misc/avptestapp/intro.vxml
Verify
Mutual Certificate Authentication: Yes No
Basic Authentication: Yes No
Speech Servers
ASR: Nuance
TTS: Nuance
Languages: English(USA) en-US
Voices: English(USA) en-US Jennifer F
Application Launch
Inbound Inbound Default Outbound
Number Number Range URI
Called Number:
Add
60011
Remove
Speech Parameters
ASR
Confidence Threshold:
Sensitivity Level:
Speed vs. Accuracy:
N Best List Length:
No Input Timeout: millisecond(s)
Recognition Timeout: millisecond(s)
Speech Complete Timeout: 800 millisecond(s)
Speech Incomplete Timeout: millisecond(s)
Maximum Grammar Cache Age: second(s)
Minimum Grammar Freshness Time: second(s)
Maximum Grammar Staleness: second(s)
Vendor Parameters:
TTS
Prosody Volume: <NONE> or
Prosody Rate: <NONE> or
Vendor Parameters:
Reporting Parameters
Advanced Parameters
Save
Cancel
Help

8.2.2. Verifying the LumenVox ASR

- Dial into the application using the Called Number specified in the newly created application.
- At the main menu, press 1 for speech recognition test.
- When prompted, speak "Open the window."
- Confirm that the application understands the utterance.

Note: For optimal results, avoid use of a speakerphone when testing the ASR, as it may introduce recognition issues.

8.3. LumenVox Test-to-Speech Server

The Avaya test application (usually installed with the MPP in **/mpp/misc/avptestapp/intro.vxml**) may be used to test the TTS. Perform the following steps:

1. Configure the test application to use the LumenVox TTS.
2. Dial into the application.
3. At the main menu, press 2 for text-to-speech test.
4. Confirm TTS speaking.

9. Conclusion

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, LumenVox Call Progress Analysis with Avaya Aura® Experience Portal. All feature and serviceability test cases were completed successfully without any observations as described in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Experience Portal*, June 2013

LumenVox help documentation, including detailed installation and configuration instructions, is available online at http://www.lumenvox.com/help/

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.